IN THE CLAIMS

1.   (currently amended) An audio signal processing method comprising the steps of:

supplying an audio signal to each of a plurality of digital filters, the digital filters corresponding to respective amplitude characteristics;

respectively supplying outputs from the plurality of digital filers to a plurality of speakers arranged in a speaker array to form a sound field;

setting a ~~predetermined~~ delay time in each of the plurality of digital filters so that transmission delay times with which the audio signal arrives at a first point in the sound field via each of the plurality of digital filters and each of the plurality of speakers will coincide with each other;

adjusting at least one amplitude characteristic of the plurality of digital filters such that the frequency response to the audio signal at a second point in the sound field is lower than the frequency response to the audio signal at the first point in the sound field, where the at least one amplitude characteristic is estimated by predicting a sample count of the signal at the second point and selecting effective ones of the amplitude characteristics corresponding to the sample count; and

adjusting cut-off frequency of a variable high pass filter ~~to filter the audio signal~~ and the delay time in each of the digital filters based on the adjusted amplitude characteristics ~~such that the frequency response to the audio signal at the second point in the sound field is suppressed relative to the frequency response to the audio signal at the first point in the sound field~~.

2.   (previously presented) The audio signal processing method according to claim 1, wherein a sound wave from the

speaker array is caused to reach at least one of the first and second points after it is reflected by a wall surface.

3.   (previously presented) The audio signal processing method according to claim 1, wherein when forming the first and second points in the sound field, a filter factor of each of the plurality of digital filters is determined by calculation and set for each of the plurality of digital filters.

4.   (original) The audio signal processing method according to claim 1, wherein when forming the first and second points in the sound field, a filter factor of each of the plurality of digital filters is read from a data base and set for each of the plurality of digital filters.

5.   (previously presented) The audio signal processing method according to claim 1, wherein:

the predetermined delay time set for at least one of the plurality of digital filters is divided into an integer part and decimal part in units of a sampling period of the audio signal;

over-sampling an impulse response including a delay time represented by at least the decimal part of the predetermined delay time for a shorter period than a sampling period to provide a sample train, wherein the sample train is down-sampled to provide pulse-waveform data of the sampling period; and

factor data is set for a part to be delayed by the plurality of digital filters based on the pulse-waveform data.

6.   (original) The audio signal processing method according to claim 5, wherein the audio signal is delayed by a part of the predetermined delay time, which is a multiple of the sampling period, by digital delay circuits which operate for the sampling period, while it is being delayed by the remainder of

the predetermined delay time, which includes the decimal part by the digital filters.

7.   (previously presented) The audio signal processing method according to claim 5, wherein:

an over-sampling period of the over-sampling operation is 1/N (N is an integer larger than or equal to 2) of the sampling period of the digital signal; and

when the delay time represented by the decimal part is nearly an integral multiple (m) of the over-sampling period, m/N is adopted as the decimal part.

8.   (previously presented) The audio signal processing method according to claim 7, wherein:

the pulse-waveform data to be delayed by a delay time which is m/N (m = 1 to N - 1) of the sampling period is pre-stored in a data base; and

pulse-waveform data approximate to the decimal part is taken out of the stored pulse-waveform data and set as a filter factor of each of the plurality of digital filters.

9.   (previously presented) The audio signal processing method according to claim 5, wherein a transfer characteristic providing a predetermined acoustic effect is convoluted in the pulse-waveform data and set as a filter factor of each of the plurality of digital filters.

10.   (currently amended) An audio signal processor comprising a plurality of digital filters, the digital filters corresponding to respective amplitude characteristics and each digital filter being supplied with an audio signal, wherein

each of the plurality of digital filters supplies an output signal to each of a plurality of speakers arranged in a speaker array to form a sound field;

each of the plurality of digital filters has a ~~predetermined~~ delay time so that transmission delay times with which the audio signal arrives at a first point in the sound field via each of the plurality of digital filers and each of the plurality of speakers will coincide with each other; and

each of the plurality of digital filters has an amplitude characteristic such that the frequency response to the audio signal at a second point in the sound field is lower than the frequency response to the audio signal at the first point in the sound field, <u>at least one amplitude characteristic is estimated by predicting a sample count of the signal at the second point and selecting effective ones of the amplitude characteristics corresponding to the sample count</u>;

<u>the audio signal is passed through a variable high pass filter, the cut-off frequency of the variable high pass filter and the delay time in each of the digital filters are adjusted based on the estimated amplitude characteristics</u> ~~that is adjusted to filter the audio signal such that the frequency response to the audio signal at the second point in the sound field is suppressed relative to the frequency response to the audio signal at the first point in the sound field~~.

11. (previously presented) The audio signal processor according to claim 10, wherein a sound wave from the speaker array is caused to reach at least one of the first and second points after it is reflected by a wall surface.

12. (previously presented) The audio signal processor according to claim 10, wherein when forming the first and second points in the sound filter, a filter factor of each of the

plurality of digital filters is determined by calculation and set for each of the plurality of digital filters.

13.   (original) The audio signal processor according to claim 10, wherein when forming the first and second points in the sound field, a filter factor o each of the plurality of digital filters is read from a data based and set for each of the plurality of digital filters.

14.   (previously presented) The audio signal processor according to claim 10, wherein:

the predetermined delay time set for at least one of the plurality of digital filters is divided into an integer part and decimal part in units of a sampling period of the audio signal,

there is further provided a calculation circuit to calculate pulse-waveform data of the sampling period by over-sampling an impulse response including a delay time represented by at least the decimal part of the predetermined delay time for a shorter period than the sampling period to provide a sample train, and down-sampling the sample train; and

the pulse-waveform provided by the calculation circuit is set as a filter factor of each of the plurality of digital filters.

15.   (previously presented) The audio signal processor according to claim 14, wherein:

an over-sampling period of the over-sampling in the calculation circuit is 1/N (N is an integer larger than or equal to 2) of the sampling period of the digital signal; and

when the delay time represented by the decimal part is nearly an integral multiple (m) of the over-sampling period, n/N is adopted as the decimal part.

16. (previously presented) The audio signal processor according to claim 14, wherein a transfer characteristic providing a predetermined acoustic effect is convoluted in the pulse-waveform data to set synthetic-waveform data as a filter factor of each of the plurality of digital filters.

17. (previously presented) The audio signal processor according to claim 10, wherein:

the predetermined delay time set for at least one of the plurality of digital filters is divided into an integer part and decimal part in units of a sampling period of the audio signal;

there is further provided a storing means for storing pulse-waveform data of the sampling period provided by over-sampling an impulse response including a delay time represented by at least the decimal part of the predetermined delay time for a shorter period than the sampling period to provide a sample train, and down-sampling the sample train; and

the pulse-waveform data stored in the storing means is taken out and set as a filter factor of each of the plurality of digital filters.

18. (previously presented) The audio signal processor according to claim 17, wherein:

an over-sampling period of the over-sampling is 1/N (N is an integer larger than or equal to 2) of the sampling period of the digital signal; and

when the delay time represented by the decimal part is nearly an integral multiple (m) of the over-sampling period, m/N is adopted as the decimal part.

19. (previously presented) The audio signal processor according to claim 17, wherein:

a plurality of the pulse-waveform data corresponding to the decimal part is pre-stored in the storing means; and

pulse-waveform data approximate to the decimal part is taken out of the stored pulse-waveform data and set as a filter factor of each of the plurality of digital filters.

20. (previously presented) The audio signal processor according to claim 17, wherein a transfer characteristic providing a predetermined acoustic effect is convoluted in the pulse-waveform data to set the pulse-waveform data as a filter factor of each of the plurality of digital filters.